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09/807,015	11/06/2001	Torsten Prange	10191/1892	8352

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EXAMINER	
OPSASNICK, MICHAEL N	

ART UNIT	PAPER NUMBER
2626	

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Please find below and/or attached an Office communication concerning this application or proceeding.

Office Action Summary	Application No. 09/807,015	Applicant(s) PRANGE ET AL.	
	Examiner Michael N. Opsasnick	Art Unit 2626	

-- *The MAILING DATE of this communication appears on the cover sheet with the correspondence address* --

Period for Reply

A SHORTENED STATUTORY PERIOD FOR REPLY IS SET TO EXPIRE 3 MONTH(S) FROM THE MAILING DATE OF THIS COMMUNICATION.

- Extensions of time may be available under the provisions of 37 CFR 1.136(a). In no event, however, may a reply be timely filed after SIX (6) MONTHS from the mailing date of this communication.
- If the period for reply specified above is less than thirty (30) days, a reply within the statutory minimum of thirty (30) days will be considered timely.
- If NO period for reply is specified above, the maximum statutory period will apply and will expire SIX (6) MONTHS from the mailing date of this communication.
- Failure to reply within the set or extended period for reply will, by statute, cause the application to become ABANDONED (35 U.S.C. § 133). Any reply received by the Office later than three months after the mailing date of this communication, even if timely filed, may reduce any earned patent term adjustment. See 37 CFR 1.704(b).

Status

1)☒ Responsive to communication(s) filed on *10 February 2006*.

2a)☒ This action is **FINAL**. 2b)☐ This action is non-final.

3)☐ Since this application is in condition for allowance except for formal matters, prosecution as to the merits is closed in accordance with the practice under *Ex parte Quayle*, 1935 C.D. 11, 453 O.G. 213.

Disposition of Claims

4)☒ Claim(s) *11-24* is/are pending in the application.

4a) Of the above claim(s) _____ is/are withdrawn from consideration.

5)☐ Claim(s) _____ is/are allowed.

6)☒ Claim(s) *11-24* is/are rejected.

7)☐ Claim(s) _____ is/are objected to.

8)☐ Claim(s) _____ are subject to restriction and/or election requirement.

Application Papers

9)☐ The specification is objected to by the Examiner.

10)☒ The drawing(s) filed on *10 February 2006* is/are: a)☒ accepted or b)☐ objected to by the Examiner.

Applicant may not request that any objection to the drawing(s) be held in abeyance. See 37 CFR 1.85(a).

Replacement drawing sheet(s) including the correction is required if the drawing(s) is objected to. See 37 CFR 1.121(d).

11)☐ The oath or declaration is objected to by the Examiner. Note the attached Office Action or form PTO-152.

Priority under 35 U.S.C. § 119

12)☒ Acknowledgment is made of a claim for foreign priority under 35 U.S.C. § 119(a)-(d) or (f).

a)☐ All b)☐ Some * c)☐ None of:

1.☐ Certified copies of the priority documents have been received.

2.☐ Certified copies of the priority documents have been received in Application No. _____.

3.☒ Copies of the certified copies of the priority documents have been received in this National Stage application from the International Bureau (PCT Rule 17.2(a)).

* See the attached detailed Office action for a list of the certified copies not received.

Attachment(s)

1) ☒ Notice of References Cited (PTO-892)

2) ☐ Notice of Draftsperson's Patent Drawing Review (PTO-948)

3) ☐ Information Disclosure Statement(s) (PTO-1449 or PTO/SB/08) Paper No(s)/Mail Date _____.

4) ☐ Interview Summary (PTO-413) Paper No(s)/Mail Date. _____ .

5) ☐ Notice of Informal Patent Application (PTO-152)

6) ☐ Other: _____.

DETAILED ACTION

Response to Amendment

1. The amendment to the title and drawings, filed 2/10/06, has been approved by the examiner.

Claim Rejections - 35 USC § 102

2. The following is a quotation of the appropriate paragraphs of 35 U.S.C. 102 that form the basis for the rejections under this section made in this Office action:

> A person shall be entitled to a patent unless –
>
> (e) the invention was described in (1) an application for patent, published under section 122(b), by another filed in the United States before the invention by the applicant for patent or (2) a patent granted on an application for patent by another filed in the United States before the invention by the applicant for patent, except that an international application filed under the treaty defined in section 351(a) shall have the effects for purposes of this subsection of an application filed in the United States only if the international application designated the United States and was published under Article 21(2) of such treaty in the English language.

3. Claims 11-19,22-24 are rejected under 35 U.S.C. 102(e) as being anticipated by Fette et al (5797121).

As per claim 11, Fette et al (5797121), teaches a method for one of coding and decoding speech sample values (as a vocoder – abstract; the encoder section – fig 4, and the decoder section – fig. 7) comprising the steps of:

"quantizing values previously obtained by an analysis from the speech signal samples values and used for a generation of speech signal parameters before being stored

in codebooks/code tables, the quantizing occurring to a word length that result in no noticeable losses in speech quality" as scalar quantizing (col. 3 lines 38-50) the already derived speech parameters based upon the input speech (col. 3 lines 24-36 -- speech parameters such as ten linear predictive speech coefficients characterized as lsf's, reflection coefficients, etc., and represented as 24 or 48 bit floating point integers), wherein the scalar quantizing transforms this 24/48 bit representation into a fixed N bit number (col. 3 lines 42-47; col. 4 lines 40-43); wherein the bit representation is chosen at the point wherein one less bit representation would degrade the quality of the representation (col. 4 lines 1-7);

"storing in the codebooks/code tables the values previously obtained by the analysis from the speech signal sample values and used for the generation of speech signal parameters" as accessing the already stored high precision speech parameters (col. 3 lines –53, referring to the database/codebook of speech parameters in col. 3 lines 24-27; Fette et al (5797121) also teaches storing the scalar quantized codebook as well -- col. 4 lines 46-49);

"scaling the values of each codebook/code table such that an available range of values is exploited as completely as possible, the step of scaling including the steps of determining a maximum of a positive value and a minimum of a negative value of each codebook/code table" as scaling according to the maximum and minimum of the speech parameters (Col. 3 lines 40-47; examiner notes that the initial speech parameter representation is in 32 bit floating point format, and that the maximum/minimum range is not limited to positive numbers only)

" if the available range of values is exceeded, performing a multiplication of the values of each codebook/code table by a first factor smaller than one, and repeating the multiplication until all elements are located in the available range of values; and causing a number of repeated multiplications to be used as a scaling factor for all codebook/code table entries" as generating a range that represents all of the high precision speech parameters (Fig. 2, subblocks 40-44) and dividing the range into 2^N intervals (Fig. 2, subblock 46) and after testing the initial quantization of the high precision parameters into an N bit representation, reducing the number of bits N to a number N-1 bits (Fig. 2, subblocks 48,50,52,54, back to subblock 48); until the minimum representation is achieved before sound quality degradation (col. 3 line 61 – col. 4 line 7). This reduction of 1 (one) bit representation is reducing the representation by a factor of 2, or 0.5. (for example, if the initial codelength is 32 bits, the number of representations for the range is 2^{32}, or 4,294,967,296 representations; and after reducing the number of bits to 31, the number of representations for the range is now 2^{31}, or 2,147,483,648, or half of the number of representations of 2^{32}).

As per claim 12, <u>Fette et al (5797121)</u> teaches the method according to claim 11 wherein the method is performed in accordance with a method of analysis through synthesis (examiner also notes that the speech parameters generated are through well know analysis by synthesis techniques – col. 3, lines 25-35 -- the linear prediction coefficients using codebooks, a type of analysis-by-synthesis coder; and col. 8 lines 13-18).

As per claim 13, Fette et al (5797121) teaches the method according to claim 11 wherein the speech quality loss/evaluation is conducted thru hearing tests (col. 3 lines 57-60).

As per claim 14, Fette et al (5797121) teaches the method according to claim 11 wherein the first factor being .5 (the reduction of 1 (one) bit representation (Fig. 2, subblock 54 feedback into subblock 48) is reducing the representation by a factor of 2, or .5; for example, if the initial codelength is 32 bits, the number of representations for the range is 2^{32}, or 4,294,967,296 representations; and after reducing the number of bits to 31, the number of representations for the range is now 2^{31}, or 2,147,483,648, or half of the number of representations of 2^{32}).

As per claim 15, Fette et al (5797121) teaches the method according to claim 11 wherein the determining word lengths of the values stored in the codebooks/code table through hearing tests (col. 3 line 55 – col. 4 line 7).

As per claim 16, Fette et al (5797121) teaches the method according to claim 11, further comprising the step of scaling the codebook/ code table entries to but of a required value range (as scaling down the range down to 2^N representation – col. 3 lines 39-45).

As per claim 17, Fette et al (5797121) teaches the method according to claim 16, further comprising the step of rounding and truncation of decimal places (as completing the scale-down of the parameters into fixed bits – col. 4 lines 40–46; examiner notes that the final storing of bit representation involves the calculation of dividing the established range for the high precision

speech parameters in col. 3 lines 40-44, dividing that range into 2^N intervals wherein N is the final determined bit representation, and converting the high precision 32 bit floating point representations into an N bit representation in col. 4 lines 44-45 (ranging from 2 bit to 5 bit representation), inherently involves rounding and truncation).

As per claim 18, Fette et al (5797121) teaches the use of a 32 bit word length (col. 4 lines 40-44; an artisan skilled in the art would readily recognize that using a 16 bit word length would only require different parameter organization while achieving equivalent results – col. 8 lines 10-12; especially since the largest quantized codevector needs 5 bits representation (fig. 3, LSF2); and that a 16 bit representation of the LSF's would only require a re-organization of the nomenclature unto itself).

As per claim 19, Fette et al (5797121) teaches the method according to claim 11 further comprising causing a processing of the codebook/code table entries to occur in accordance with a digital signal processing in a whole-number format (as the codebook entries represented initially in 32 bit floating point number representation – col. 4 lines 38-41).

As per claim 22, Fette et al (5797121) teaches an apparatus corresponding to one of a coder and a decoder for processing speech signal sample values in accordance with a method of analysis through synthesis (as a vocoder – abstract; the encoder section – fig 4, and the decoder section – fig. 7; examiner also notes that the speech parameters generated are through well know

analysis by synthesis techniques – col. 3, lines 25-35 -- the linear prediction coefficients using codebooks, a type of analysis-by-synthesis coder; col. 8 lines 13-18) comprising:

"an arrangement for storing in quantized form values contained in codebooks/code tables for a generation of speech signal parameters; an arrangement for selecting a word length such that no noticeable losses in speech quality occur; an arrangement for quantizing the values contained in the codebooks/code tables to the word length that results in no noticeable losses in speech quality" as scalar quantizing (col. 3 lines 38-50) the already derived speech parameters based upon the input speech (col. 3 lines 24-36 -- speech parameters such as ten linear predictive speech coefficients characterized as lsf's, reflection coefficients, etc., and represented as 24 or 48 bit floating point integers), wherein the scalar quantizing transforms this 24/48 bit representation into a fixed N bit number (col. 3 lines 42-47; col. 4 lines 40-43); wherein the bit representation is chosen at the point wherein one less bit representation would degrade the quality of the representation (col. 4 lines 1-7);

"an arrangement for scaling the values of each codebook/code table such that an available range of values can be exploited as completely as possible; an arrangement for determining a maximum of positive values and negative values of each codebook/code table" as scaling according to the maximum and minimum of the speech parameters (Col. 3 lines 40-47; examiner notes that the initial speech parameter representation is in 32 bit floating point format, and that the maximum/minimum range is not limited to positive numbers only)

"and for multiplying the values of each codebook/code table by a first factor less than one if the if the available range of values is exceeded; and an arrangement for, if a multiplication of the values of the codebooks/code tables lies outside the available range of values, performing

repeated multiplication until all elements are located in the available range of values, and for providing a number of repeated multiplication as a scaling factor" as generating a range that represents all of the high precision speech parameters (Fig. 2, subblocks 40-44) and dividing the range into 2^N intervals (Fig. 2, subblock 46) and after testing the initial quantization of the high precision parameters into an N bit representation, reducing the number of bits N to a number N-1 bits (Fig. 2, subblocks 48,50,52,54, back to subblock 48); until the minimum representation is achieved before sound quality degradation (col. 3 line 61 – col. 4 line 7). This reduction of 1 (one) bit representation is reducing the representation by a factor of 2, or .5. (for example, if the initial codelength is 32 bits, the number of representations for the range is 2^{32}, or 4,294,967,296 representations; and after reducing the number of bits to 31, the number of representations for the range is now 2^{31}, or 2,147,483,648, or half of the number of representations of 2^{32}).

As per claim 23, Fette et al (5797121) teaches the apparatus according to claim 22, wherein the noticeable losses in speech quality are determined through a hearing test (col. 3 lines 56-60).

As per claim 24, Fette et al (5797121) teaches the apparatus according to claim 22 wherein the first factor is 0.5 ((the reduction of 1 (one) bit representation (Fig. 2, subblock 54 feedback into subblock 48) is reducing the representation by a factor of 2, or .5; for example, if the initial codelength is 32 bits, the number of representations for the range is 2^{32}, or 4,294,967,296 representations; and after reducing the number of bits to 31, the number of

representations for the range is now 2^{31}, or 2,147,483,648, or half of the number of representations of 2^{32}).

Claim Rejections - 35 USC § 103

4. The following is a quotation of 35 U.S.C. 103(a) which forms the basis for all obviousness rejections set forth in this Office action:

> (a) A patent may not be obtained though the invention is not identically disclosed or described as set forth in section 102 of this title, if the differences between the subject matter sought to be patented and the prior art are such that the subject matter as a whole would have been obvious at the time the invention was made to a person having ordinary skill in the art to which said subject matter pertains. Patentability shall not be negatived by the manner in which the invention was made.

5. Claim 20 is rejected under 35 U.S.C. 103(a) as being unpatentable over Fette et al (5797121) in further view of Gersho et al (6233550).

As per claim 20, Fette et al (5797121) teaches the method according to claim 11, as shown above. Fette et al (5797121) discusses the use of his scalar design to be applied to vector codebooks (col. 4, lines 15-25 containing speech parameters including lines spectral frequencies, cepstral coefficients (both in col. 3 lines 25-32), along with other well-known speech coders (col. 8 lines 13-16). Fette et al (5797121) does not explicitly teach the claimed harmonic vector coding with spectral envelopes and LPCs; however, Gersho et al (6233550) teach a Harmonic coder (Fig. 4a) wherein in LP residuals, and spectral magnitudes are quantized (Fig. 4b, Fig. 4c; fig. 11). Therefore, it would have been obvious to one of ordinary skill in the art of speech coders/decoders at the time the invention was made to use the quantization procedure of Fette et al (5797121) into the different vector quantized parameters in Gersho et al

(6233550) because it would introduce the saving of bits (i.e., using less bits giving the same representation of speech, as discussed above in Fette et al (5797121)), wherein such an efficiency being useful in harmonic coders (Gersho et al (6233550), col. 4 lines 58-65).

6. Claim 21 is rejected under 35 U.S.C. 103(a) as being unpatentable over Fette et al (5797121) in further view of Tzeng (5307441).

As per claim 21, Fette et al (5797121) teaches the method according to claim 11, as shown above. Fette et al (5797121) discusses the use of his scalar design to be applied to vector codebooks (col. 4, lines 15-25 containing speech parameters including lines spectral frequencies, cepstral coefficients (both in col. 3 lines 25-32), along with other well-know speech coders (col. 8 lines 13-16). Fette et al (5797121) does not explicitly teaches the claimed CELP, values for LSP, VQ vector quantization, as well as gain VQ's; however, Tzeng teaches the use of quantization in an analysis by synthesis type coder (Fig 1; col. 2 lines 63-67) wherein the LSP, VQ's and gain VQ's are quantized (Tzeng (5307441), col. 12, lines 58-65; col. 7 lines 40-53). Therefore, it would have been obvious to one of ordinary skill in the art of speech coders/decoders at the time the invention was made to use the quantization procedure of Fette et al (5797121) into the different vector quantized parameters in Tzeng (5307441) because it would advantageously reduce memory requirements (Tzeng (5307441), col. 2 lines 65-68) as well as improving performance without memory overload, as well as saving data bits (col. 3, lines 1-15).

Response to Arguments

7. Applicant's arguments filed 2/10/2006 have been fully considered but they are not persuasive. As per applicant's arguments toward the claim scope of "no noticeable losses in speech quality", examiner argues that the claim scope is, in itself, subjective in interpretation. The applied passage of Fette, "do not yield significant speech quality degradation", or in other words, "no significant speech quality degradation", is clearly toward the concept of speech quality, and that the question becomes the differential, if any, between "noticeable" and "significant". Turning to Webster's II New College Dictionary, we see that for the definition of "noticeable", the word "significant" is used as a possible definition. Therefore, the two words are synonymous, and therefore, the Fette reference teaches the claim scope of "no noticeable losses in speech quality". As per applicant's arguments towards scaling, examiner respectfully disagrees and argues that the 32 bit floating point of Fette is not limited to positive numbers only, and therefore covers the range of positive and negative numbers.

Conclusion

8. **THIS ACTION IS MADE FINAL.** Applicant is reminded of the extension of time policy as set forth in 37 CFR 1.136(a).

A shortened statutory period for reply to this final action is set to expire THREE MONTHS from the mailing date of this action. In the event a first reply is filed within TWO MONTHS of the mailing date of this final action and the advisory action is not mailed until after

the end of the THREE-MONTH shortened statutory period, then the shortened statutory period will expire on the date the advisory action is mailed, and any extension fee pursuant to 37 CFR 1.136(a) will be calculated from the mailing date of the advisory action. In no event, however, will the statutory period for reply expire later than SIX MONTHS from the mailing date of this final action.

9. Any inquiry concerning this communication or earlier communications from the examiner should be directed to Michael Opsasnick, telephone number (571)272-7623, who is available Tuesday-Thursday, 9am-4pm.

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, Mr. Richemond Dorvil, can be reached at (571)272-7602. The fax phone number for the organization where this application or proceeding is assigned is 571-273-8300.

Information regarding the status of an application may be obtained from the Patent Application Information Retrieval (PAIR) system. Status information for published applications may be obtained from either Private PAIR or Public PAIR. Status information for unpublished applications is available through Private PAIR only. For more information about the PAIR system, see http://pair-direct.uspto.gov. Should you have questions on access to the Private PAIR system, contact the Electronic Business Center (EBC) at 866-217-9197 (toll-free).

mno
5/10/06

Michael N. Opsasnick
Examiner
Art Unit 2626